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Lab Report # 02
Digital Signal Processing

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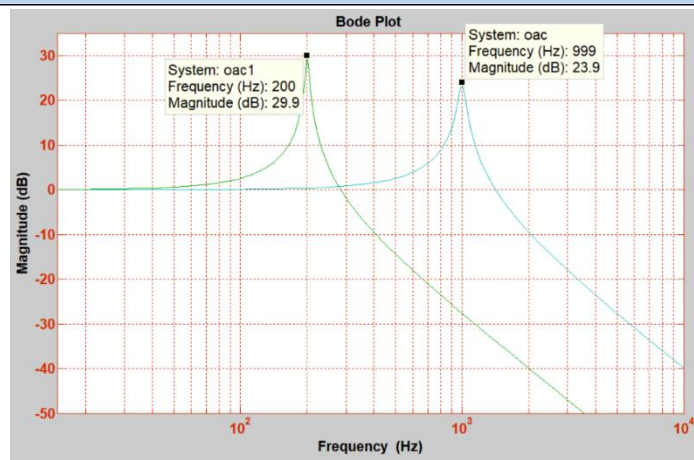
Task-1

The RC filter circuit is implemented on MATLAB by using the given impulse response.
The "**audioread**" command is used to read the given audio file. The impulse response is cut after **500 samples** (0-499) and then convolved with the given impulse response by using the "**conv**" command. The sampling rate is determined as **44,100 Hz.**
Effect of Filter: The filtered sound is heard via laptop's speaker using **"sound"** command. The sound of the guitar was muffled because the filter has removed some higher frequency content

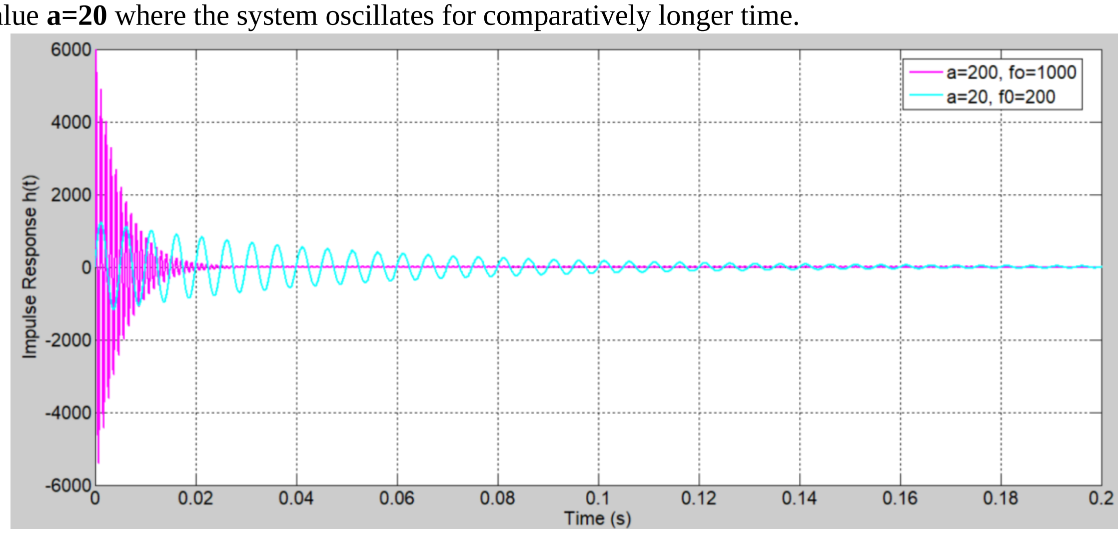
Task-2

The transfer function of the given frequency response is written on MATLAB using **"tf"** command. The frequency response is plotted using the **"bode**" command for two different values of the given parameters (a, f_0). The frequency response (magnitudes) are compared in the adjacent graph.
It is observed that for **a=200, f_0=1000**, the cut of frequency is **999 Hz** and amplitude is **23.9 dB**. And **for a=20, f_0=200**, the cut of frequency is **200 Hz** and amplitude is **29.9 dB**. So, the bode plot shows that **f_0** is the cut-off frequency of the filter.



Task-3

Similarly, the given impulse response of the filter is plotted for both the given parameters. Like step 1, the sample period is defined by time vector "`T = [0:1: 10000]*Ts;` " and the impulse response is plotted for time T but different parameters (a, f_0). It is observed that for higher value of damping factor **a=200**, the system is damped earlier compared to the lower value **a=20** where the system oscillates for comparatively longer time.



Task-4

Finally, the guitar signal is filtered through the oscillatory circuit and the output is heard via laptop speaker is for botthe different values of a, f0. After cutting the impulse response to 1000 samples the impulse response is convolved with the guitar signal. A high-volume sound is heard for a=200, f0 = 1000 compared to a lower muffled sound for a=20, f0 = 200. This is because the for f0 = 1000 the filter is allowing the frequencies upto 1000Hz (like a low pass filter) to pass through. And for f0 = 200 the filter is only allowing the frequencies upto 200Hz to pass though it is resulting and blocking the frequencies above 200 that results in lower sound or volume

References

[1] Lecture Notes from Digital Signals Processing, WS2020 by Prof. Dr. Stefan Brückl.
[2] Laboratory Handouts from Digital Signals Processing by Prof. Marcus Maresch.
[3] Online: www.mathworks.com. Accessed on 25-Nov-2020.